Amendments to the Claims

Please amend Claims 1, 26, 32 and 57. Please add new Claim 62. The Claim Listing below will replace all prior versions of the claims in the application:

Claim Listing

1. (Currently Amended) In a communications system for transmitting a near end digital signal using a compression code comprising a plurality of parameters including a first parameter, said parameters representing an audio signal comprising a plurality of audio characteristics, said compression code being decodable by a plurality of decoding steps, said communications system also transmitting a far end digital signal using a compression code, apparatus for reducing echo in said near end digital signal comprising:

~~a first handset generating a near end digital signal using a compression code comprising a plurality of parameters including a first parameter,~~

~~wherein said parameters represent an audio signal comprising a plurality of audio characteristics,~~

~~wherein said compression code is decodable by a plurality of decoding steps,~~

~~a second handset generating a far end digital signal using a compression code; and~~

a processor responsive to said near end digital signal to read at least said first parameter of said plurality of parameters, to perform at least one of said plurality of decoding steps on said near end digital signal and said far end digital signal to generate at least partially decoded near end signals and at least partially decoded far end signals, and

~~wherein said partially decoded near end signals represent a first subframe of data and wherein said partially decoded far end signals represent a second subframe of data,~~

~~wherein said processor estimates an echo likelihood based on said first subframe of data and said second subframe of data,~~

~~wherein said processor, when said estimation of echo likelihood indicates an echo, is~~ responsive to said at least partially decoded near end signals and at least partially decoded far end signals to adjust said first parameter to generate an adjusted first parameter and to replace at least said first parameter with said adjusted first parameter in said near end digital signal to reduce echo in the near end digital signal.

2. (Original) Apparatus, as claimed in claim 1, wherein said first parameter is a quantized first parameter and wherein said processor generates said adjusted first parameter in part by quantizing said adjusted first parameter before writing said adjusted first parameter into said near end digital signal.

3. (Original) Apparatus, as claimed in claim 1, wherein said processor is responsive to said at least partially decoded near end signals and said at least partially decoded far end signals to generate an echo likelihood signal representing the amount of echo present in said partially decoded near end signals, and wherein said processor is responsive to said echo likelihood signal to adjust said first parameter.

4. (Original) Apparatus, as claimed in claim 3, wherein said characteristics comprise spectral shape and wherein said first parameter comprises a representation of filter coefficients, and wherein said processor is responsive to said echo likelihood signal to adjust said representation of filter coefficients towards a magnitude frequency response.

5. (Original) Apparatus, as claimed in claim 4, wherein said representation of filter coefficients comprises line spectral frequencies.

6. (Original) Apparatus, as claimed in claim 4, wherein said representation of filter coefficients comprises log area ratios.

7. (Original) Apparatus, as claimed in claim 4, wherein said magnitude frequency response corresponds to background noise.

8. (Original) Apparatus, as claimed in claim 1, wherein said characteristics comprise the overall level of said audio signal and wherein said first parameter comprises codebook gain.

9. (Original) Apparatus, as claimed in claim 1, wherein said first parameter comprises a codebook vector parameter.

10. (Original) Apparatus, as claimed in claim 1, wherein said characteristics comprise period of long-term correlation and wherein said first parameter comprises a pitch period parameter.

11. (Original) Apparatus, as claimed in claim 1, wherein said characteristics comprise strength of long-term correlation and wherein said first parameter comprises a pitch gain parameter.

12. (Original) Apparatus, as claimed in claim 1, wherein said characteristics comprise spectral shape and wherein said first parameter comprises a representation of filter coefficients.

13. (Original) Apparatus, as claimed in claim 12, wherein said representation of filter coefficients comprises log area ratios.

14. (Original) Apparatus, as claimed in claim 12, wherein said representation of filter coefficients comprises line spectral frequencies.

15. (Original) Apparatus, as claimed in claim 12, wherein said representation of filter coefficients corresponds to a linear predictive coding synthesis filter.

16. (Original) Apparatus, as claimed in claim 1, wherein said first parameter corresponds to a first characteristic of said plurality of audio characteristics, wherein said plurality of decoding steps comprises at least one decoding step avoiding substantial altering of said first characteristic and wherein said processor avoids performing said at least one decoding step.

17. (Original) Apparatus, as claimed in claim 16, wherein said audio characteristic comprises power and wherein said first characteristic comprises power.

18. (Original) Apparatus, as claimed in claim 16, wherein said at least one decoding step comprises post-filtering.

19. (Original) Apparatus, as claimed in claim 1, wherein said compression code comprises a linear predictive code.

20. (Original) Apparatus, as claimed in claim1, wherein said compression code comprises regular pulse excitation – long term prediction code.

21. (Original) Apparatus, as claimed in claim1, wherein said compression code comprises code-excited linear prediction code.

22. (Original) Apparatus, as claimed in claim 1, wherein said first parameter comprises a series of first parameters received over time, wherein said processor is responsive to said near end digital signal to read said series of first parameters, and wherein said processor is responsive to said at least partially decoded near end and far end signals and to at least a plurality of said series of first parameters to generate said adjusted first parameter.

23. (Original) Apparatus, as claimed in claim 1, wherein said compression code is arranged in frames of said digital signals and wherein said frames comprise a plurality of subframes each comprising said first parameter, wherein said processor is responsive to said compression code to read at least said first parameter from each of said plurality of subframes, and wherein said processor replaces said first parameter with said adjusted first parameter in each of said plurality of subframes.

24. (Original) Apparatus, as claimed in claim 23, wherein said processor reads said first parameter from a first of said subframes, begins to perform at least a plurality of said decoding steps on said near end digital signal during said first subframe and replaces said first parameter with said adjusted first parameter before processing a subframe following the first subframe so as to achieve lower delay.

25. (Original) Apparatus, as claimed in claim 1, wherein said compression code is arranged in frames of said digital signals and wherein said frames comprise a plurality of subframes each comprising said first parameter, wherein said processor performs at least a plurality of said decoding steps during a first of said subframes to generate said at least partially decoded near end and far end signals, reads said first parameter from a second of said subframes occurring subsequent to said first subframe, generates said adjusted first parameter in response to said at least partially decoded near end and far end signals and said first parameter, and replaces said first parameter of said second subframe with said adjusted first parameter.

26. (Currently Amended) In a communications system for transmitting a near end digital signal comprising code samples, said code samples comprising first bits using a compression code and second bits using a linear code, said code samples representing an audio signal, said audio signal having a plurality of audio characteristics, said system also transmitting a far end digital signal, apparatus for reducing echo comprising:

~~a first handset transmitting a near end digital signal comprising code samples,~~

~~wherein said code samples comprise first bits using an compression code and second bits using a linear code,~~

~~wherein said code samples represent an audio signal,~~

~~wherein said audio signal has a plurality of audio characteristics;~~

~~a second handset transmitting a far end digital signal,~~

~~wherein said near end digital signals and said far end digital signals each represent a subframe of data; and~~

~~a processor, wherein said processor partially decodes said near end and said far end digital signals for use in determining an echo estimate,~~

~~wherein said processor, when said echo estimate indicates the presence of an echo, is~~ a processor responsive to said near end digital signal and said far end digital signal to adjust said first bits and said second bits , without decoding said compression code in said near end digital signal, to reduce echo in the near end digital signal; and

a transmitter to transmit the first and second bits in an adjusted state to a far end device to present the first and second bits in an audible form to an end user.

27. (Canceled)

28. (Original) Apparatus, as claimed in claim 26, wherein said linear code comprises pulse code modulation (PCM) code.

29. (Original) Apparatus, as claimed in claim 26, wherein said compression code samples conform to the tandem-free operation of the global system for mobile communications standard.

30. (Original) Apparatus, as claimed in claim 26, wherein said first bits comprise the two least significant bits of said samples and wherein said second bits comprise the 6 most significant bits of said samples.

31. (Original) Apparatus, as claimed in claim 29, wherein said 6 most significant bits comprise PCM code.

32. (Currently Amended) In a communications system for transmitting a near end digital signal using a compression code comprising a plurality of parameters including a first parameter, said parameters representing an audio signal comprising a plurality of audio characteristics, said compression code being decodable by a plurality of decoding steps, said communications system also transmitting a far end digital signal using a compression code, a method of reducing echo in said near end digital signal comprising:

~~receiving a near end digital signal using a compression code comprising a plurality of parameters including a first parameter,~~

~~wherein said parameters represent an audio signal comprising a plurality of audio characteristics,~~

~~wherein said compression code is decodable by a plurality of decoding steps;~~

~~receiving a far end digital signal using a compression code, wherein said near end digital signal and said far end digital signal represent a subframe of data; and~~

reading at least said first parameter of said plurality of parameters in response to said near end digital signal;

performing at least one of said plurality of decoding steps on said near end digital signal and said far end digital signal to generate at least partially decoded near end signals and at least partially decoded far end signals;

~~determining the presence of an echo using said partially decoded near end signals and said partially decoded far end signals;~~

adjusting[[,]] ~~when an echo is present,~~ said first parameter in response to said at least partially decoded near end signals and at least partially decoded far end signals to generate an adjusted first parameter; and

replacing at least said first parameter with said adjusted first parameter in said near end digital signal to reduce echo in the near end digital signal.

33. (Original) A method, as claimed in claim 31, wherein said first parameter is a quantized first parameter and wherein said adjusting comprises generating said adjusted first parameter in part by quantizing said adjusted first parameter.

34. (Original) A method, as claimed in claim 31, wherein said adjusting comprises generating an echo likelihood signal representing the amount of echo present in said partially decoded near end signals in response to said at least partially decoded near end signals and said at least partially decoded far end signals, and wherein said adjusting further comprises adjusting said first parameter in response to said echo likelihood signal.

35. (Original) A method, as claimed in claim 33, wherein said characteristics comprise spectral shape and wherein said first parameter comprises a representation of filter coefficients, and

wherein said adjusting comprises adjusting said representation of filter coefficients towards a magnitude frequency response in response to said echo likelihood signal.

36. (Original) A method, as claimed in claim 34, wherein said representation of filter coefficients comprises line spectral frequencies.

37. (Original) A method, as claimed in claim 34, wherein said representation of filter coefficients comprises log area ratios.

38. (Original) A method, as claimed in claim 34, wherein said magnitude frequency response corresponds to background noise.

39. (Original) A method, as claimed in claim 31, wherein said characteristics comprise the overall level of said audio signal and wherein said first parameter comprises codebook gain.

40. (Original) A method, as claimed in claim 31, wherein said first parameter comprises a codebook vector parameter.

41. (Original) A method, as claimed in claim 31, wherein said characteristics comprise period of long-term correlation and wherein said first parameter comprises a pitch period parameter.

42. (Original) A method, as claimed in claim 31, wherein said characteristics comprise strength of long-term correlation and wherein said first parameter comprises a pitch gain parameter.

43. (Original) A method, as claimed in claim 31, wherein said characteristics comprise spectral shape and wherein said first parameter comprises a representation of filter coefficients.

44. (Original) A method, as claimed in claim 42, wherein said representation of filter coefficients comprises log area ratios.

45. (Original) A method, as claimed in claim 42, wherein said representation of filter coefficients comprises line spectral frequencies.

46. (Original) A method, as claimed in claim 42, wherein said representation of filter coefficients corresponds to a linear predictive coding synthesis filter.

47. (Original) A method, as claimed in claim 31, wherein said first parameter corresponds to a first characteristic of said plurality of audio characteristics, wherein said plurality of decoding steps comprises at least one decoding step avoiding substantial altering of said first characteristic and wherein said performing at least a plurality of said decoding steps comprises avoiding performing said at least one decoding step.

48. (Original) A method, as claimed in claim 46, wherein said audio characteristic comprises power and wherein said first characteristic comprises power.

49. (Original) A method, as claimed in claim 46, wherein said at least one decoding step comprises post-filtering.

50. (Original) A method, as claimed in claim 31, wherein said compression code comprises a linear predictive code.

51. (Original) A method, as claimed in claim 31, wherein said compression code comprises regular pulse excitation – long term prediction code.

52. (Original) A method, as claimed in claim 31, wherein said compression code comprises code-excited linear prediction code.

53. (Original) A method, as claimed in claim 31, wherein said first parameter comprises a series of first parameters received over time, wherein said reading comprises reading said series of first parameters, and wherein said adjusting comprises generating said adjusted first parameter in response to said at least partially decoded near end and far end signals and to at least a plurality of said series of first parameters.

54. (Original) A method, as claimed in claim 31, wherein said compression code is arranged in frames of said digital signals and wherein said frames comprise a plurality of subframes each comprising said first parameter, wherein said reading comprises reading at least said first parameter from each of said plurality of subframes in response to said compression code, and wherein said

replacing comprises replacing said first parameter with said adjusted first parameter in each of said plurality of subframes.

55. (Original) A method, as claimed in claim 53, wherein said reading comprises reading said first parameter from a first of said subframes, wherein said performing comprises beginning to perform at least a plurality of said decoding steps on said near end digital signal during said first subframe and wherein said replacing comprises replacing said first parameter with said adjusted first parameter before processing a subframe following the first subframe so as to achieve lower delay.

56. (Original) A method, as claimed in claim 31, wherein said compression code is arranged in frames of said digital signals and wherein said frames comprise a plurality of subframes each comprising said first parameter, wherein said performing comprises performing at least a plurality of said decoding steps during a first of said subframes to generate said at least partially decoded near end and far end signals, wherein said reading comprises reading said first parameter from a second of said subframes occurring subsequent to said first subframe, wherein said adjusting comprises generating said adjusted first parameter in response to said at least partially decoded near end and far end signals and said first parameter, and wherein said replacing comprises replacing said first parameter of said second subframe with said adjusted first parameter.

57. (Currently Amended) In a communications system <u>for transmitting a near end digital signal comprising code samples, said code samples comprising first bits using a compression code and second bits using a linear code, said code samples representing an audio signal, said audio</u>

signal having a plurality of audio characteristics, said system also transmitting a far end digital signal, a method of reducing echo in said near end digital signal, comprising:

~~receiving a near end digital signal comprising code samples, wherein said code samples comprise first bits using a compression code and second bits using a linear code,~~

~~wherein said code samples representing an audio signal,~~

~~wherein said audio signal has a plurality of audio characteristics;~~

~~receiving a far end digital signal, wherein said near end digital signal and said far end digital signal each represent a subframe of data;~~

~~partially decoding said near end digital signal to form a partially decoded near end digital signal;~~

~~partially decoding said far end digital signal to form a partially decoded far end digital signal;~~

~~determining the presence of an echo based on said partially decoded near end digital signal and said partially decoded far end digital signal; and~~

adjusting[[,]] ~~when an echo is present,~~ said first bits and said second bits, without decoding said compression code in said near end digital signal, in response to said near end digital signal and said far end digital signal[[,]] ~~in order~~ to reduce ~~control the~~ echo characteristics of said near-end digital signal; and

transmitting the first and second bits in an adjusted state to a far end device to present the first and second bits in audible form to an end user.

58. (Original) A method, as claimed in claim 56, wherein said linear code comprises pulse code modulation (PCM) code.

59. (Original) A method, as claimed in claim 56, wherein said compression code samples conform to the tandem-free operation of the global system for mobile communications standard.

60. (Original) A method, as claimed in claim 56, wherein said first bits comprise the two least significant bits of said samples and wherein said second bits comprise the 6 most significant bits of said samples.

61. (Original) A method, as claimed in claim 59, wherein said 6 most significant bits comprise PCM code.

62. (New) Apparatus for reducing echo in a coded domain signal, comprising:

a near end partial decoder to at least partially decode coded near end digital signals, including at least a first parameter of a plurality of parameters representing respective near end audio signals in the coded near end digital signals to form at least partially decoded near end signals;

a far end partial decoder to at least partially decode coded far end digital signals, including at least a first parameter of a plurality of parameters representing respective far end audio signals in the coded far end digital signals to form at least partially decoded far end signals;

a processor responsive to said near end digital signals to read at least said first parameter of first said plurality of parameters in the coded near end digital signals and at least partially decode said near end digital signal and to read a coded far end digital signal to generate at least partially decoded far end signals and at least partially decoded far end signals, and responsive to at least said partially decoded near end signals and at least partially decoded far end signals to

adjust said first parameter to generate an adjusted first parameter and to replace at least said first parameter with said adjusted first parameter in said near end digital signal to reduce echo in the near end digital signal.